EXHIBIT A

MICROELECTRONICS PATENT COMMITTEE INVENTION SUBMISSION

Name(s) of Submitters	Telephone No:	Loc/Room	SBU/Org. Title:	HR ID:	E-Mail Address
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IP LAW USE
Submission No:____________
Date Received:____________
Attorney:____________

TITLE: An optimal method for sharing data channel with voice signals

Important Notes: (1) Keep in mind that your submission should be written so it can be understood in 5 to 10 minutes by a generalist. Avoid the use of undefined acronyms and jargon. Keep the language simple. (2) Have any of the above submitter(s) discussed this invention with, or provided an invention submission disclosing this invention to, an attorney other than the recipient of this invention submission? ____YES _X_NO

1. **Describe the problem your invention solves:**
   Optimally transmit voice signals through a data channel.

2. **Based on information of which you are already aware, describe:**
   (i) previous attempts to solve the problem your invention solves:
   None.

   (ii) the disadvantages of the previous attempts:
   N/A.

3. **Summarize (30 words or less) the new feature(s) of your invention that solve the problem:**
   Optimally load voice bits onto a data channel according to the corresponding SNR gap.

4. **Succinctly describe your invention, referring to drawings, sketches, photographs, etc., in sufficient detail to enable one knowledgeable in the invention's field of technology to understand construction and operation of the invention. Drawings, etc., should show only those features necessary for an understanding of the invention. Describe how/why your invention overcomes the disadvantages noted in 2. (ii) above.**
   See the attached memo "An optimal method for sharing data channel with voice signals".

5. **Advantages of your invention:**
   Transmit more bits with same channel.

6. **Explain how use of your invention would be detected:**
   The invention is intended to propose to national and international standard committees to be included into the standards.

*** Provide the information requested in this box on each page of the submission, as well as drawings, sketches, photographs, etc. ***

Submitter(s) signature(s) and date:

____Jingdong Lin__________ ______ date

____________________ ______ date

This invention submission has been read and understood by the following two witnesses:

____________________ ______ date

An Optimal Method for Sharing Data Channel with Voice Signals

1. Background

With the development of the wide band telecommunication systems, using data channel to carry voice signals becomes more and more popular in various telecommunication systems. From voice over DSL, voice over cable modem, voice over LAN (wireline or wireless) to voice over home networks (Home PNA in particular), the merge of the data and voice transmissions forms a significant trend.

However, in the current development, as long as we know, all these systems handle the voice bits the same way as they handle the data bits. The advantage of doing this is the simplicity of the system. However, on the other hand, by doing this, a remarkable portion of the channel bandwidth will be wasted.

2. Theory

The quality requirements on the data and voice bits are significantly different. Usually, the BER of a data bit need to be as low as 10^{-7}, while the BER requirements on a voice bit is just 10^{-3}. Thus, using a data channel to transmit the voice signal in the similar way as data is over-killing and can waste a lot of precious bandwidth.

Theoretically, the channel capacity can be determined by the following famous formula originally derived by Shannon.

$$C = \log_2(1 + \frac{S}{N})$$

Where S/N is the signal to noise ratio.

However, for a certain system with a particular type of modulation scheme, there is a gap between the practically achievable channel capacity and the theoretical channel capacity. Take the QAM modulation as an example, the channel capacity is

$$C = \log_2(1 + \frac{3}{\Delta^2} \cdot \frac{S}{N})$$

The inverse of additional term $\Delta^2/3$ is called the SNR gap, or simply gap.

The gap is a function of required BER p. The relation can be expressed in the formula below.

$$p = \frac{2}{\sqrt{2\pi}} \int_{\Delta}^{\infty} e^{-\frac{x^2}{2}} dx$$

For often used BER values, the corresponding values of Δ and gap are listed in the table below.

BER	10^{-2}	10^{-3}	10^{-4}	10^{-5}	10^{-6}	10^{-7}
Δ	2.577	3.287	3.891	4.418	4.892	5.333
Gap (dB)	3.45	5.56	7.03	8.13	9.02	9.77

The difference in the SNR gap will determine how many bits you can load to a symbol of QAM or other modulations.

3. The Method of Optimal Loading

From the above derivations, the best place to begin treating the data and voice differently is the bit to symbol mapping unit. Here, the data and voice bits are loaded to the symbols, and the difference in their BER requirements can be fully employed.

To do this, the symbols have to be separated into two groups, one group is used to load data bits and the other group is used to load voice bits. Different modulation constellations will be used for each group. A denser constellation (more bits per symbol) will be used for the data bits, while a less dense constellation will be used for voice bits. The exact number of bits per symbol for either the data bits or the voice bits can be calculated according to the available signal to noise ratio and the BER requirements of the bits. By doing this, the different BER requirements of either the data bits or the voice bits can be guaranteed while the efficient bandwidth usage is achieved in an optimal way.

Also take QAM as an example, the number of bits to be loaded to a QAM symbol when SNR is know is usually calculated as

$$b = \log_2(1 + 10^{\frac{G}{10}})$$

Where

$$G = \text{SNR} - \text{gap} - \text{margin} + \text{CodingGain}$$

From the formula, more bits can be loaded if gap is smaller.

The separation of the data and voice symbols can be accomplished by time, by frequency, by coding etc. according to the characteristics of a particular system.

4. Applied the Method to a Multi-carrier System

A multi-carrier system can be G.Lite, G.DMT, DMT based VDSL, IEEE 802.11 based system, Home-Plug based home network system, etc.

When the method is applied to a DMT system, the separation of the data and voice bits can be achieved by frequency. In another word, all the tones (sun-carriers) can be divided into two parts. One part will be loaded with voice bits by using the SNR gap

corresponding to high BER (10^{-3}), while the other part will be loaded with data bits by using the SNR gap of low BER (10^{-7}).

One particular way to separate the tones can be described below:

- Arrange all the tones in the order of measured signal to noise ratio from low to high, for the tones with similar SNR values, the one with the lower index goes first;
- Load the voice bits first from the tones with the low SNR;
- After the voice bits are loaded, begin to load the data bits from the next unloaded tone;
- Make necessary communications between TX and RX side to share the information of voice and data separation.

However, the separate transmission of the data bits and voice bits can also be achieved in many other ways.

5. Applied the Method to a Single Carrier System

A single carrier system can be RADSL, CAP/QAM based VDSL, HDSL, Home networks (Home PNA or Home Plug), or IEEE 802.16 based systems, etc.

When the method is applied to a CAP/QAM system, the separation of the data and voice bits can be achieved by time. In another word, the data bits and voice bits can be organized into different time bursts and being transmitted alternatively. The symbols in the data burst will be loaded with a denser constellation, while the symbols in the voice burst will be loaded with a less dense constellation.

However, the separate transmission of the data bits and voice bits can also be achieved in many other ways.

6. An Example

An example is provided below on a DMT system. Similar examples can also be demonstrated for QAM based systems.

For a G.Lite (G.992.2) based system, the downstream transmission is considered. It is assumed that the signal is transmitted via the loop model of T1.601 number 2. The interference is 24 self NEXT and 24 self FEXT. It is also assumed that

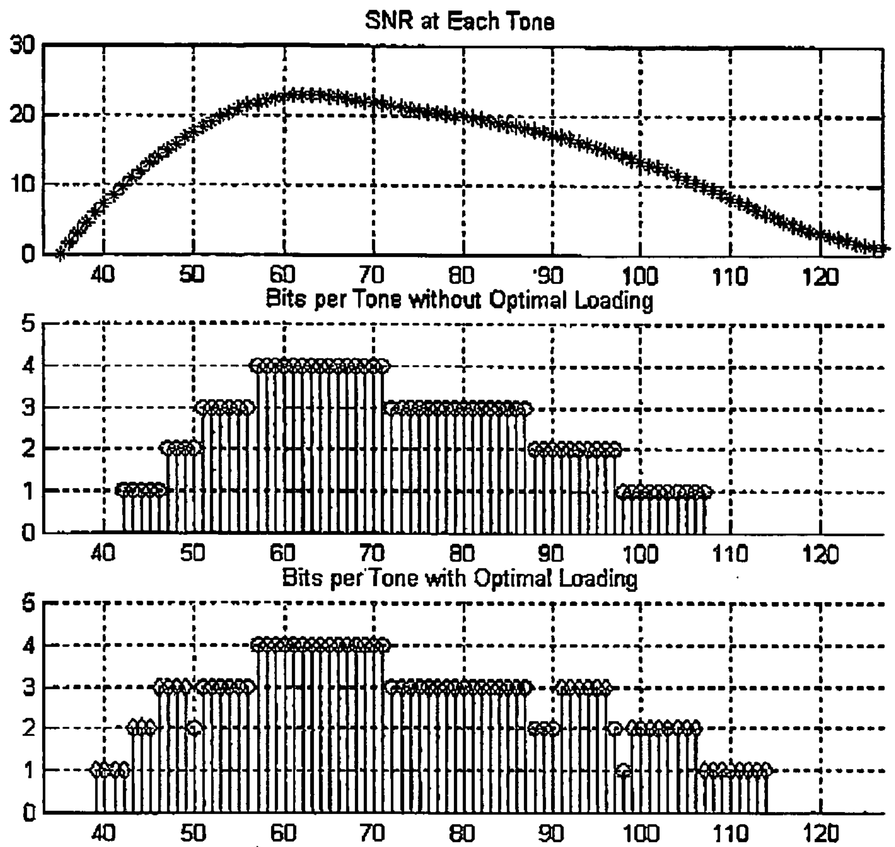
$$\text{Margin} = \text{CodingGain} = 6 \text{ dB}$$

If voice and data bits are not separated and all being loaded to sub-carriers 33-128 based on the SNR gap of 9.77 dB (BER 10^{-7}), 169 bits can be transmitted. With the symbol rate of 4 ksps, the bit rate is 676 kbps. If 256 kbps need to be transmitted, the available bandwidth for data bits is 420 kbps.

Optimal Bit Loading

However, if the voice bits are separated from the data bits and loaded with the SNR gap of 5.56 dB by using the optimal loading mother we described in the previous sections, the remaining channel bandwidth left for data will be 548 kbps. It is 128 kbps more.

The figure below demonstrated the example.



In the figure, the top one is the SNR (in dB) of each sub-carrier, from number 33 to number 128. The middle one shows the results of the conventional loading. The bottom one shows the results of the optimal loading with voice and data bits separated. The red diamond terminated tones are loaded with the voice bits while the blue circle terminated tones are loaded with data bits.

It can be observed by comparing the middle and the bottom figures that some low SNR tones at two ends will be wasted by the conventional loading method. In this particular example, it stands for 10 additional bits or 40 kbps for the voice signal.

It is also can be observed from the bottom figure that usually one more bits can be added to each tone if the voice bits are loaded instead of the data bits.